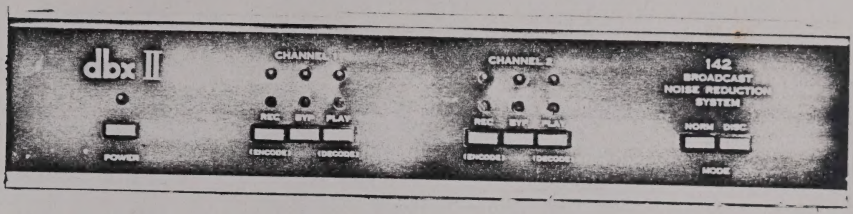
142 MANUAL

dbx II

INSTRUCTION



MANUAL

DBX 142

BROADCAST NOISE REDUCTION SYSTEM

INSTRUCTION MANUAL

TABLE OF CONTENTS

	PAGE
SECTION I: DESCRIPTION	2
What makes the dbx system unique among noise reduction systems	3
What is compression/expansion? A brief discussion of compander theory	3
SECTION II: SPECIFICATIONS	6
SECTION III: CONTROLS, CONNECTORS & INSTALLATION	7
Front panel features	7
Rear panel features	8
Installation notes:	
Power mains	9
Mounting	9
Level and impedance	9
Audio and remote bypass connections	9
Preparation of a jumper connector	10
Installation block diagrams:	
Typical 2-channel record & playback installations	11
2-channel installation for remote lines or microwave	12
Installation for monaural recording & playback	13
Installation for playback of dbx-encoded discs	14
Installation for noise reduction of time delay units	15
How to make a transfer of encoded tapes or discs	16
SECTION IV: ADJUSTMENTS	
Level match calibration	17
Tape Machine alignment	18
SECTION V: OPERATION	
Hints for optimum utilization of the 142	20
Theory of operation:	21
Which method of signal level detection is used?	21
Evolution of the dbx system design	21
Tape noise...where it comes from	22
Using signal pre-emphasis and de-emphasis	23
Using pre-emphasis of the level sensor	23
Using RMS detection to avoid phase sensitivity	24
Using bandpass filters to avoid mistracking	26
System performance	27
SECTION VI: APPENDIX	28
Schematic diagrams	
Assembly drawings	
Warranty/service information	

SECTION I
DESCRIPTION

The dbx Model 142 Broadcast Noise Reduction System is a two-channel compressor/expander providing a minimum of 30dB broadband audio noise reduction with a 10dB headroom improvement in the storage or transmission medium. The two channels are independent and the front panel is switchable to record (encode), play (decode) or bypass functions. In addition, either channel may be remotely bypassed.

In the record mode, the system compresses the input signal by a 2:1 ratio, linear (with respect to decibels) over a 100dB range (refer to figure 1). Upon playback, the circuitry is switched to provide 1:2 expansion that is a mirror image of the compression (encoding). The result is a tape recording that cannot be distinguished from the original audio source.

dbx II noise reduction, as used in the Model 142, is a sophisticated application of the classical compression/expansion concept of noise reduction. It is identical to the original dbx professional recording studio system, except for minor differences in the level detection circuitry and in the pre-emphasis/de-emphasis curves. While both dbx systems provide the same amount of noise reduction and headroom improvement, dbx II is further optimized to meet the specific requirements of broadcast applications. Refer to the "Theory of Operation" section of this manual for additional information concerning dbx II.

The dbx system eliminates tape hiss that would otherwise be introduced by the recording process. The only noise heard on playback is that noise which was present in the original recorded source. For recording of live announcements, studio productions, or taped interviews, the playback will sound as "live" as the event itself. For copying existing tapes and records, the copy will sound just like the original recording, with no additional noise; dbx II does not remove noise present in the original signal. The result of dbx encoding is to minimize the audible quality differences between cart playback (historically the noisiest link in the broadcast chain) and live announce or other dbx-assisted cart playback units. The phenomenal increase in dynamic range and absence of tape-induced noise that can be achieved through dbx encode/decode processing allows an automated station to approach live sound quality.

Non-cartridge related uses for the dbx 142 system include improving the signal-to-noise ratio of older reel-to-reel recorders, eliminating hiss from TV audio tracks, and cleaning up noisy full-frequency land lines or microwave links from the studio to the transmitter. The dbx Model 142 may also be used as a playback-only device to decode dbx-encoded discs or tapes for broadcast.

A modest investment in dbx Model 142 noise reduction units can protect a much larger investment by prolonging the useful life of major studio equipment and of installations that might not otherwise meet contemporary performance requirements. Equally important, the improvement in signal quality will undoubtedly please performers, engineers and listeners who enjoy top quality audio.

WHAT MAKES THE DBX SYSTEM UNIQUE AMONG NOISE REDUCTION SYSTEMS

Many attempts have been made to apply the classical compression/expansion concept to professional audio, but most of these have been less-than-successful for technical reasons. The dbx approach, however, is unique. A voltage controlled amplifier (VCA) is used as the gain control element, and a true RMS level sensing detector is used to insure perfect encode/decode tracking for accurate transient response...irrespective of phase shifts in the transmission or storage medium. Moreover, there is no audible breathing, pumping, or other coloration of the sound.

Pilot tones and routine calibration are not necessary because there is no threshold at which compression and expansion factors change. The system will compress and expand on a linear basis throughout a range of over 100dB. Critical level matching is not needed, and the system will accurately encode and decode despite differences in record and playback levels. However, primarily for monitoring convenience, a specific level may be set as the unity gain point, using level match controls. The level chosen should correspond to your individual 0 VU standard, be that +4dBm, +8dBm, or any other value from -10 to +18dBm. In fact, the major restricting elements in choosing a nominal operating level are the headroom (maximum output capability) and the noise floor of the expander and of those circuits which follow it (refer to "B" in Figure 1).

WHAT IS COMPRESSION/EXPANSION? A BRIEF DISCUSSION OF COMPANDER THEORY

A compander, such as the dbx Model 142, is comprised of a compressor and an expander. The audio signal is compressed 2:1 prior to tape recording to decrease the dynamic range of the original signal. For example, a 100dB program (which may have peaks at +20dBm and quiet portions as low as -80dBm) after compression will be squeezed to a 50dB dynamic range. Assuming a nominal operating level of 0 dBm, the compressed program will peak at +10dBm, and will have a minimum level of -40dBm (refer to Figure 1). This allows the original signal to better conform to the usable dynamic range of the recorder by keeping the quietest signal above the noise and the doudest signal below the level of tape saturation. (The principle is the same for closed-loop transmission or for recording.)

Upon playback, the recorded signal is expanded 1:2 to precisely recreate the dynamic range of the original program material. Since the audio has been kept above any noise throughout the recording/playback (or transmission) process, the only noise present is that which was in the original program.

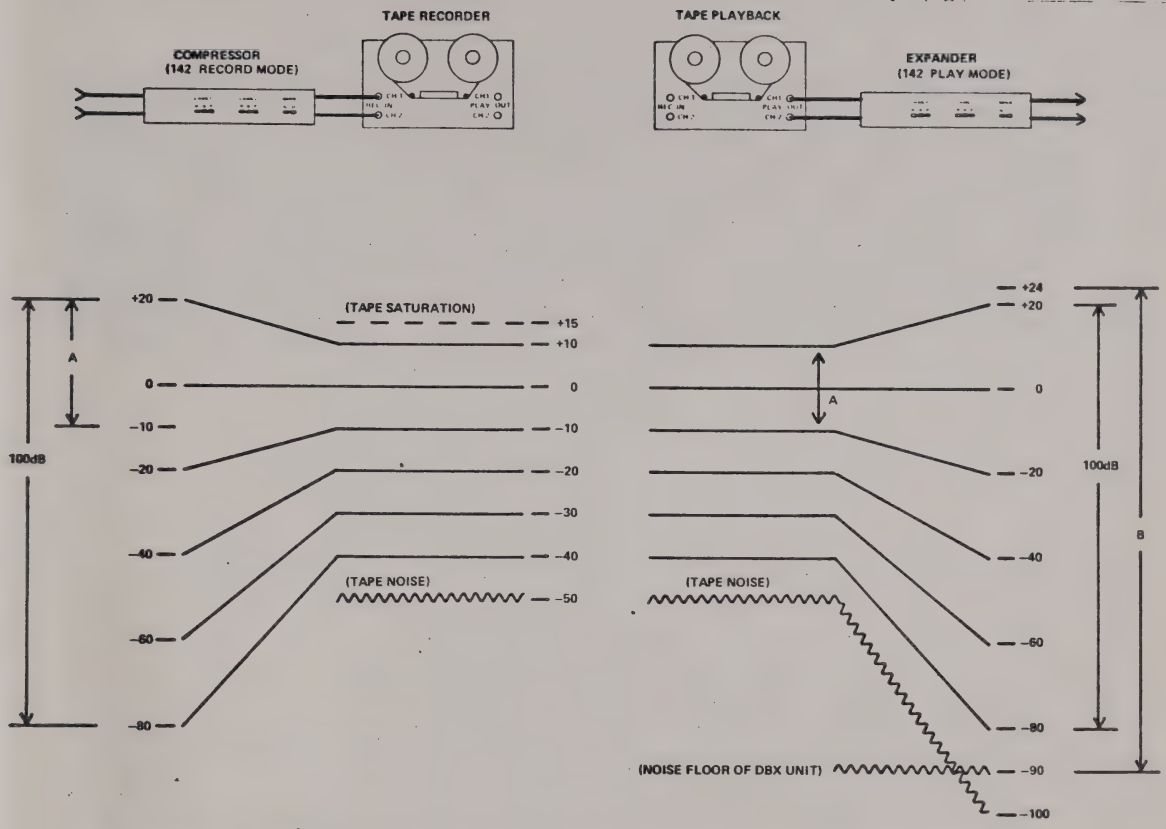


FIGURE 1. HOW A COMPRESSION/EXPANSION SYSTEM ACHIEVES NOISE REDUCTION.
(A) Nominal input level range (Record Level Match & Play Level Match) over which the unity gain point can be adjusted. Level match is actually uncritical, ±10dB (approximately).
(B) Dynamic range of dbx expander, over 100dB.

For a compander to function properly, the compressor and expander must operate as exact mirror images. Compression and expansion operations are each controlled by a signal detector, and it is dbx's exclusive RMS detectors that make possible the high degree of accuracy throughout the encode/decode cycle... regardless of phase shifts in the recording or other audio processing. Further details of dbx's RMS detection, as contrasted to peak and average detection, are presented in the "Theory of Operation" section of this manual.

In addition to applications involving tape storage (including tape delay), dbx noise reduction can be used for noise reduction in transmission lines, echo chamber send and return lines, and microwave links. [dbx-encoded signals are not recommended for general broadcast due to compatibility considerations.] In all these applications, the basic theory is the same; compress the input to

the medium, and expand the output from the medium. This approach has been used successfully to make dbx-encoded discs, the beginning of a new era in record enjoyment. When decoded on the Model 142, or any type II dbx system, these discs are free of vinyl-grain-induced surface noise. Moreover, such discs can be cut at lower levels, despite their exceptional dynamic range, so transient response and trackability are substantially improved. Another advantage of the encoded disc is that more program material can be recorded on each side. The DISC switch on the Model 142 must be engaged when playing encoded discs, but cannot be used for conventional discs. The switch changes the low frequency cutoff of the signal and of the detector, thereby preventing any record warp or rumble from creating mistracking in the expansion process.

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SECTION II
SPECIFICATIONS

INPUT IMPEDANCE	20K-ohms, balanced differential input; 10K-ohms, balanced transformer-isolated input using optional plug-in transformers. 620-ohms when rear-panel termination switch is engaged, with or without transformer option.
OUTPUT IMPEDANCE	600-ohms nominal. Actual source impedance is 80-ohms, so unit will operate into any load greater than 150-ohms.
INPUT LEVEL: MAXIMUM	+24dBm.
NOMINAL	-10dBm to +18dBm, adj. w/ Record Level Match trimmer.
OUTPUT LEVEL: MAXIMUM	+24dBm into 600-ohm termination, +30dBm into 10K-ohm or greater impedance.
NOMINAL	-10dBm to +18dBm, adj. w/Play Level Match trimmer.
BYPASS	Front panel switch provides hard-wired bypass; Remote bypass provides electronic unity gain condition, adjustable ±3dB with Bypass Level Match trimmer to compensate for possible loading effects.
DISTORTION (THD)	0.1% typical, 0.2% max. for full compress/expand cycle.
EQUIVALENT INPUT NOISE	-90dBm typical.
FREQUENCY RESPONSE	±1dB, 30Hz to 20kHz for full compress/expand cycle. -1 dB at 30Hz (NORM mode), -3 dB at 30Hz (DISC mode).
EFFECTIVE NOISE REDUCTION	Better than 30dB broadband noise reduction, with a 10dB simultaneous improvement in headroom (dbx II curve).
OPERATING TEMPERATURE	From 32°F to 140°F (0°C to 60°C).
DIMENSIONS	3½" H x 19" W x 10" D (89mm x 483mm x 254mm); for standard relay rack mounting.
NET WEIGHT	8 pounds (3.6 kg).
SHIPPING WEIGHT	12 pounds (5.5 kg).
POWER MAINS	120V AC, 50-60Hz standard (240V optional, conversion in seconds by means of rear-panel plug-in card).
POWER CONSUMPTION	10 Watts.
OPTIONAL TRANSFORMERS	Ampex #4580200-01, or equivalent; 10K:10K-ohms, octal plug-in base. One transformer per channel handles record and play inputs for that channel.

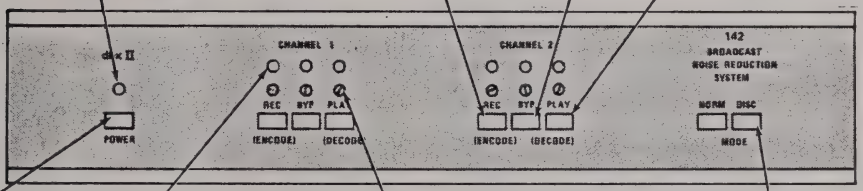
SECTION III
CONTROLS, CONNECTIONS AND INSTALLATION INSTRUCTIONS

POWER INDICATOR
LED glows red when AC power is on.

RECORD (ENCODE) SWITCH
Sets the channel for compression (encode) when recording or when driving remote feeds.

BYPASS SWITCH
Sets the channel for hard-wired bypass (no expansion or compression). This is used when noise reduction is not needed, or when transferring previously encoded program material.

PLAY (DECODE) SWITCH
Sets the channel for expansion (decode) when playing encoded tape or disc, or when receiving an encoded remote feed.



POWER SWITCH
Turns on the electronics.

CHANNEL INDICATORS
LED's glow red to indicate whether the channel is in record, play or bypass mode.

LEVEL MATCH CONTROLS
Recessed, screwdriver-adjustable potentiometers for achieving unity gain through the 142 for your nominal reference level.

Bypass level matching is for remote bypass mode only, which is not a hard-wire bypass, and may require slight trim depending on source and load impedances. Level matching is not critical for accurate encode/decode operation.

MODE SWITCHES
Set the signal and detector bandpass filters for playback of encoded discs (DISC mode) or any other encoded material (NORM mode). For recording (encode) or bypass operation, these switches have no effect.

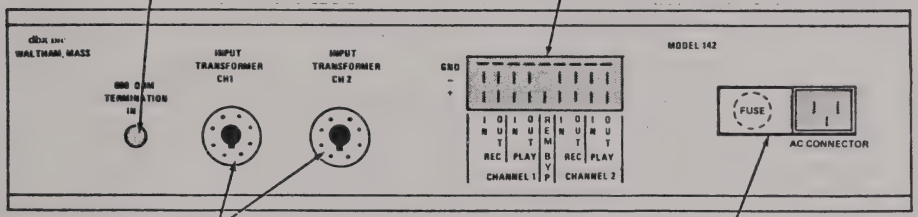
FRONT PANEL FEATURES

INPUT TERMINATION SWITCH

Depressing this switch places a 620-ohm load across the 142 record and play inputs; for use with any equipment requiring a nominal 600-ohm termination. (Switch out, the input impedance is 20K-ohms, or 10K-ohms with optional transformer.). The switch is black when not engaged, and day-glo-colored with 600-ohm termination engaged.

INPUT/OUTPUT CONNECTOR

For all audio inputs and outputs, and for remote bypass connections.



TRANSFORMER SOCKETS

For optional input transformers. The 142 input is balanced by means of a differential amplifier, and the nominal input impedance is 20K-ohms. However, where complete line isolation is required, 10K:10K-ohm transformers may be inserted in these sockets. Automatic re-routing of the input signal occurs as each transformer is installed. Only one transformer per channel is needed, and internal switching assigns it to the record input in record mode or the play input in play mode.

AC POWER INTERFACE

Includes AC fuse and voltage-selector circuit card, hidden behind sliding window, and AC connector for power cord supplied with the unit. 120V AC operation is standard, but cards are available which re-wire the power transformer primary for 100V, 220V, and 240V. Use 1/4-A AGC fuse for 120V, 1/8-A AGC fuse for 240V.

REAR PANEL FEATURES

INSTALLATION NOTES

POWER MAINS:

The dbx Model 142 is wired for 120V AC, 50-60Hz operation, as supplied. Units may be quickly modified for 100V, 220V or 240V operation by exchanging a small printed circuit card which is located behind the sliding window in the AC connector interface (on the 142 rear panel). For safety, the voltage cannot be changed unless the AC cord has been disconnected. Be sure to use the correct AC cord and mains voltage for the particular voltage-selector card. Power consumption is 10 watts.

MOUNTING:

The 142 may be mounted in a standard relay rack, with no special ventilation. However, mounting above any source of heat or strong electro-magnetic fields, such as those generated by power amplifiers or vacuum tube equipment, should be avoided. Mounting dimensions are 3½" (9cm) high x 19" (49cm) wide x 10" (25½cm) deep; mounting brackets are an integral part of the 142 cabinet.

LEVEL AND IMPEDANCE:

The 142 record and play inputs are 20KΩ, balanced by differential amplifiers. Depressing the rear-panel "600 Ohm Termination" button changes the inputs to 600Ω (balanced). Where complete input isolation is required, optional plug-in bridging transformers may be installed in the sockets provided. (Use Ampex transformer, P/N 4580200-01, or equivalent.) A unique switching arrangement provides an internal bypass of each transformer socket until a transformer is inserted; then signal is automatically re-routed through the transformer. If desired, the "600 Ohm Termination" button may also be used in conjunction with the plug-in transformers to achieve a terminated, isolated input. Nominal level is +4dBm (adjustable from -10 to +18dBm) with a maximum input level of +24dBm.

The 142 record and play outputs are designed to feed a 600Ω nominal impedance. They are balanced, transformer-isolated outputs with an actual source impedance of 80Ω, so the unit will drive any impedance greater than 150Ω. Output level is nominal +4dBm (adjustable from -10 to +18dBm), +24dBm maximum when 600Ω terminated, +30dBm maximum into 10KΩ or greater impedances.

AUDIO AND REMOTE BYPASS CONNECTIONS:

All audio input and output circuits, plus remote bypass circuits, are brought to a single input/output connector on the 142 rear panel. Figure 2, below, illustrates the connector layout, as seen on the actual rear panel. (Figure 3, next page, is an expanded view of the wiring side of the mating connector, illustrating the pin numbers and corresponding connections.)

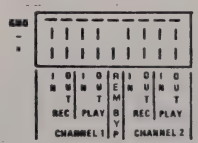


FIGURE 2. 142 INPUT/OUTPUT CONNECTOR.

Two-conductor shielded cable, such as Belden 8451, is suggested for audio connections. A single length of this same cable may be used for remote bypass connections for both channels, or the two channels can be bypassed together using a single-conductor shielded cable.

Remote bypass is achieved individually on channel 1 and on channel 2 by grounding the corresponding input/output connector pins: pin 15 for channel 1, and pin 13 for channel 2. Any of the audio ground (GND) terminals may be used as the ground return for remote bypass. Pins 13 and 15 may be tied together if both channels are always to be bypassed at the same time.

The 142 may be used with unbalanced equipment. Jumper the (-) terminal to the (GND) terminal of each channel in the input/output connector; then connect the ground side of the interconnecting cables to the (GND) terminals, and the high side of the cables to the (+) terminals.

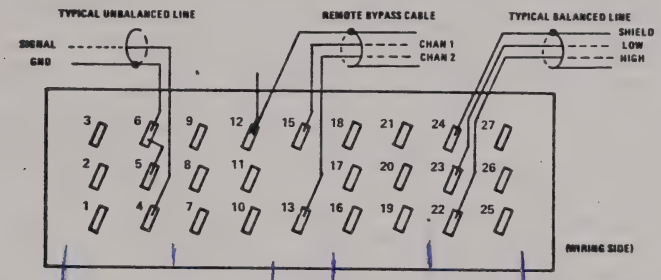


FIGURE 3. MATING CONNECTOR FOR INPUT/OUTPUT CABLES.
(TYPICAL CABLE CONNECTIONS SHOWN)

PREPARATION OF A JUMPER CONNECTOR:

We have assumed that production work and on-the-air work are most often done in different rooms. Therefore, the use of more than one dbx 142 is suggested. If your needs dictate moving a single 142 from one location to another, it may be advantageous to prepare a jumper connector. This connector may then be mated to the input/output cable when the 142 is removed, providing a simple and effective hard-wire bypass.

Use a Vernitron plug (#P-3327-04-CCT) or a Cinch Jones plug (#P-327-CCT), and wire it as illustrated in figure 4.

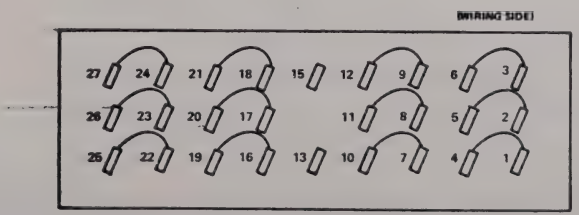


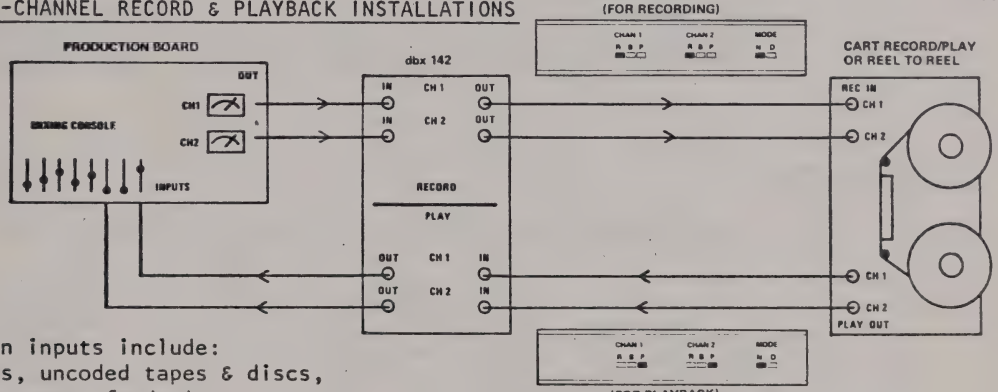
FIGURE 4. JUMPER CONNECTOR FOR CONVENIENT BYPASS WHEN 142 IS REMOVED.

INSTALLATION BLOCK DIAGRAMS

In the following diagrams, we have included front-panel representations to illustrate how the various switches should be set during different operations. A darkened switch in the illustration corresponds to a depressed button on the actual 142 front panel. Except when playing (decoding) dbx-encoded discs, the MODE switches (NORM & DISC) have no effect. Therefore, the NORM switch is shown engaged in all examples except those where DISC mode *must* be used. We have illustrated the connections to the 142 with a stylized representation of the rear panel; this is done for clarity. Refer to the rear panel artwork, or to figure 3 (p.9) for actual connections.

NOTE: Due to the almost infinite variety in studio layouts and in production techniques, we can offer no standard way to install the dbx 142. Instead, we present several examples of "typical" installations. These examples are intended only as suggestions to help you establish the best installation for your specific needs.

TYPICAL 2-CHANNEL RECORD & PLAYBACK INSTALLATIONS



Production inputs include:
live mikes, uncoded tapes & discs,
& uncoded remote feeds.*

FIGURE 5A. PRODUCTION ROOM SET-UP FOR RECORDING & PLAYBACK OF DBX-ENCODED TAPES.

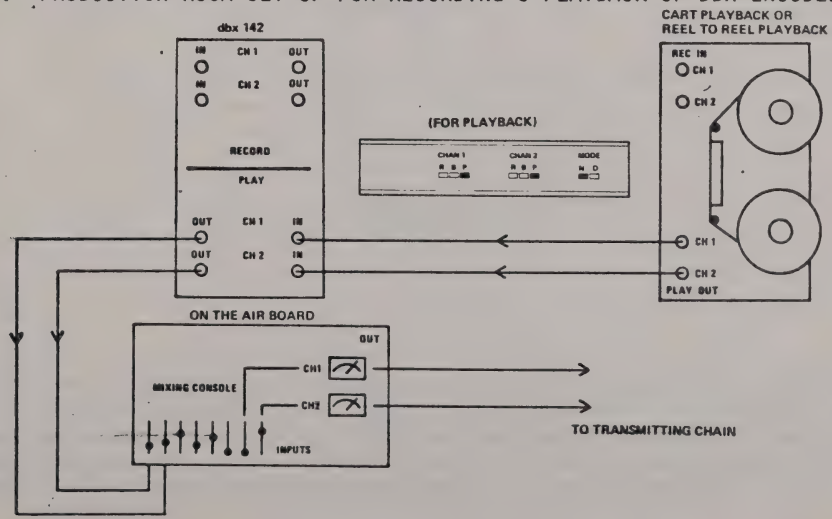


FIGURE 5B. ON-THE-AIR SET-UP FOR PLAYBACK OF DBX-ENCODED TAPES.

*Any dbx-encoded program must first be decoded, using another 142, prior to mixing. It may then be re-encoded after mixing and/or signal processing has been completed.

TYPICAL 2-CHANNEL INSTALLATION FOR REMOTE LINES OR MICROWAVE LINK

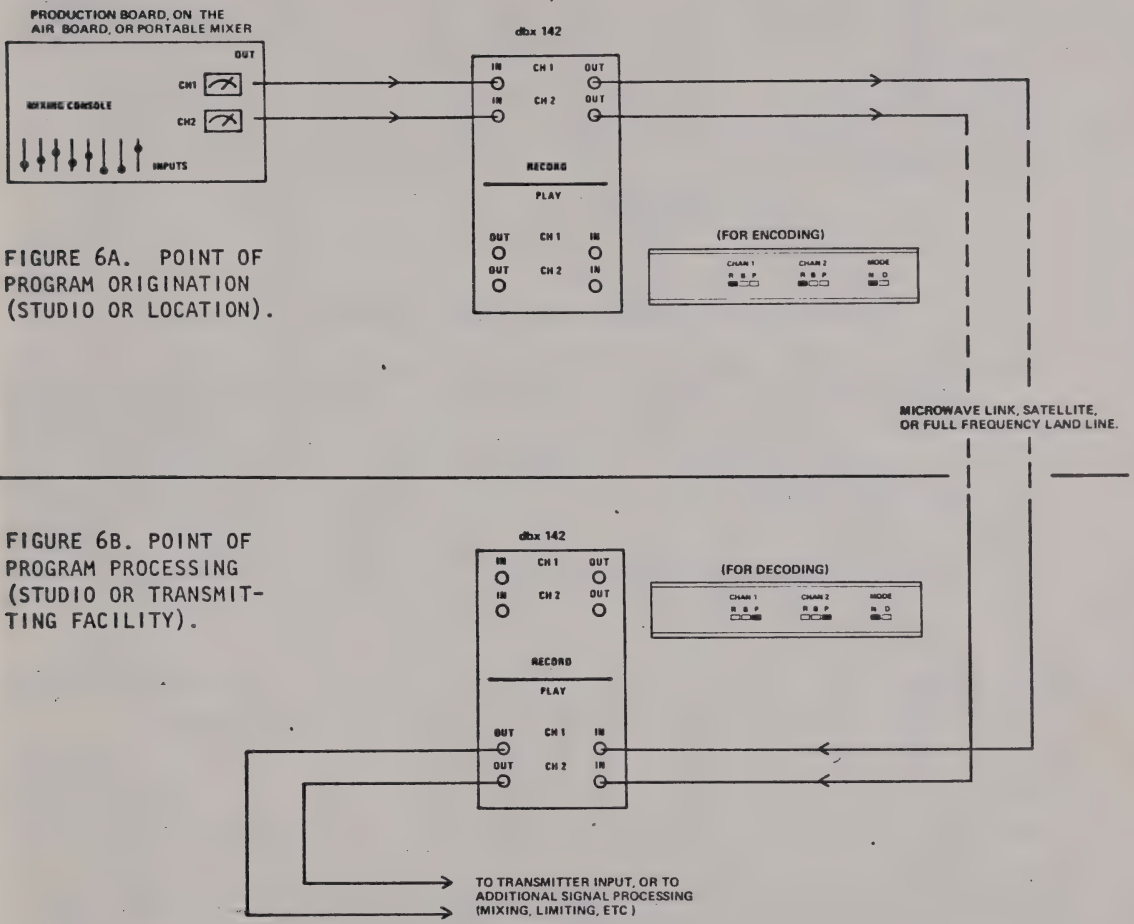


FIGURE 6A. POINT OF PROGRAM ORIGINATION (STUDIO OR LOCATION).

FIGURE 6B. POINT OF PROGRAM PROCESSING (STUDIO OR TRANSMITTING FACILITY).

NOTE: The on-the-air console's output may be encoded, relayed to the transmitter in compressed (encoded) form, then decoded just prior to being applied to the transmitter input. No limiting or additional compression whatsoever should be applied to the encoded signal; this may be done either at the studio before encoding, or at the transmitter after decoding.

TYPICAL INSTALLATION FOR MONAURAL RECORDING AND PLAYBACK

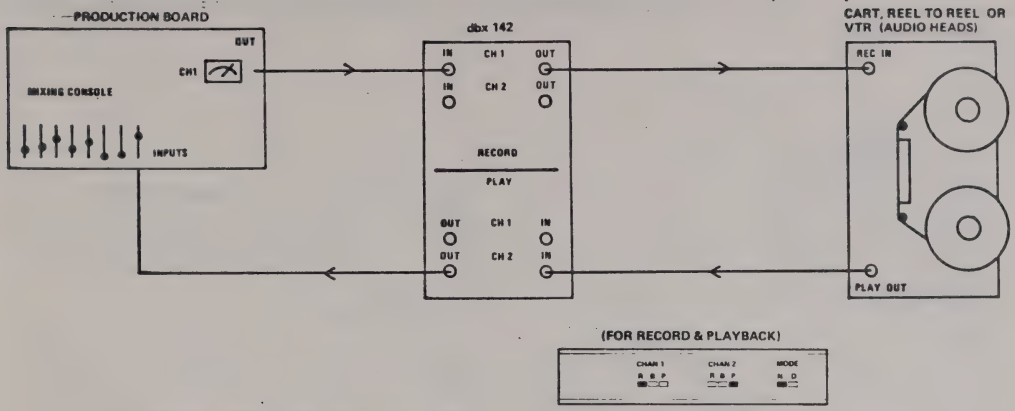


FIGURE 7A. RECORDER/PLAYER WITHOUT SEPARATE MONITOR OUTPUT

NOTE: This arrangement may be used to simultaneously encode while recording and decode the tape monitor output (for off-tape monitoring). Note that channel 1 is used for encoding, while channel 2 is used for decoding, so no switching of the 142 is necessary when changing from recording to playing modes.

With machines having separate playback and tape monitor outputs, the monitor output would be connected to channel 2, and the playback output would be connected to channel 1 of the 142 play input. In this way, after the recording is completed, normal playback decoding may be done by switching channel 1 from record to play mode (as shown below). The equivalent 2-channel instllation, having simultaneous encode/decode capability, requires two 142's.

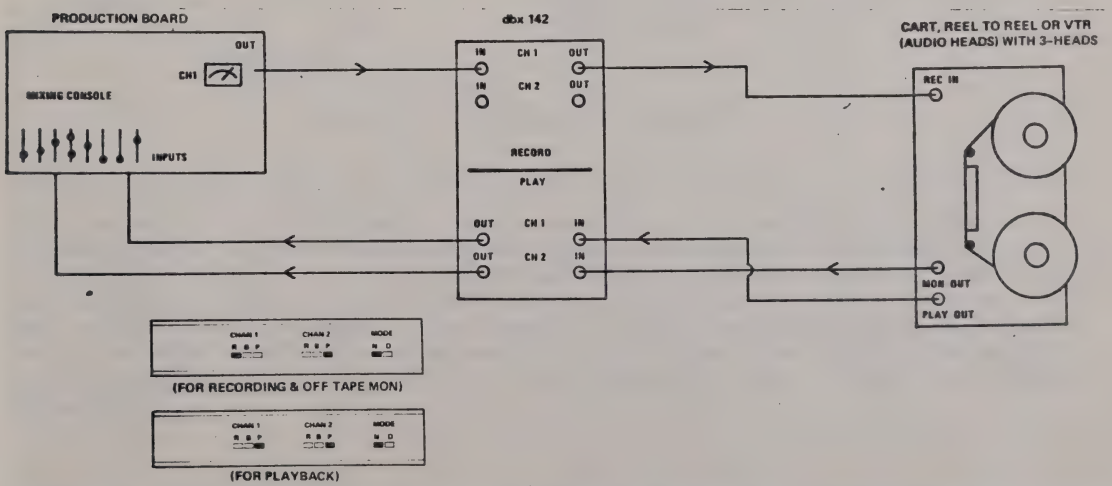


FIGURE 7B. RECORDER/PLAYER WITH OFF-TAPE MONITOR OUTPUT AND SEPARATE PLAYBACK OUTPUT.

TYPICAL INSTALLATION FOR PLAYBACK OF DBX-ENCODED DISCS

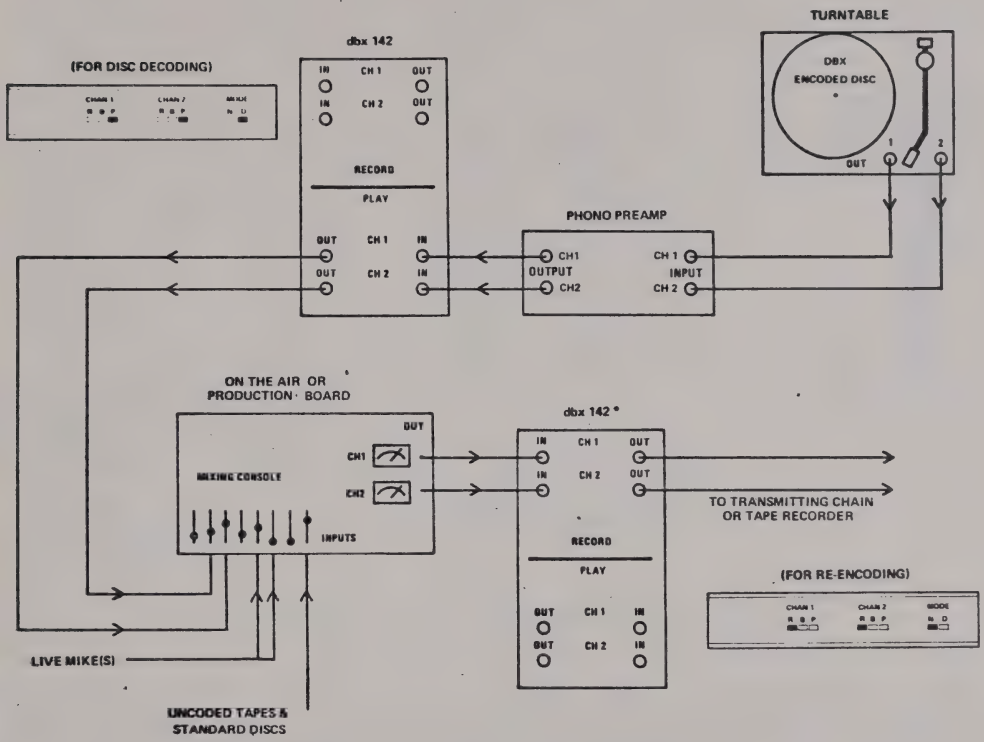


FIGURE 8. ENCODED DISC PLAYBACK

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*This second dbx 142 is optional, for re-encoding of the decoded disc. Due to possible rumble and warp components that may be introduced by disc playback, it is sometimes necessary to first DISC decode before transfer to tape; if signal flows to another dbx 142 which is set for NORM decoding, the DISC may first have to be DISC decoded, then re-encoded. (For example, a 142 at the transmitter would always be set for NORM decoding, so an encoded disc would have to be DISC decoded at the studio, then re-encoded before feed to the transmitter...even in real-time playback situations.)

TYPICAL INSTALLATION FOR NOISE REDUCTION OF REVERB, ECHO OR DELAY DEVICES

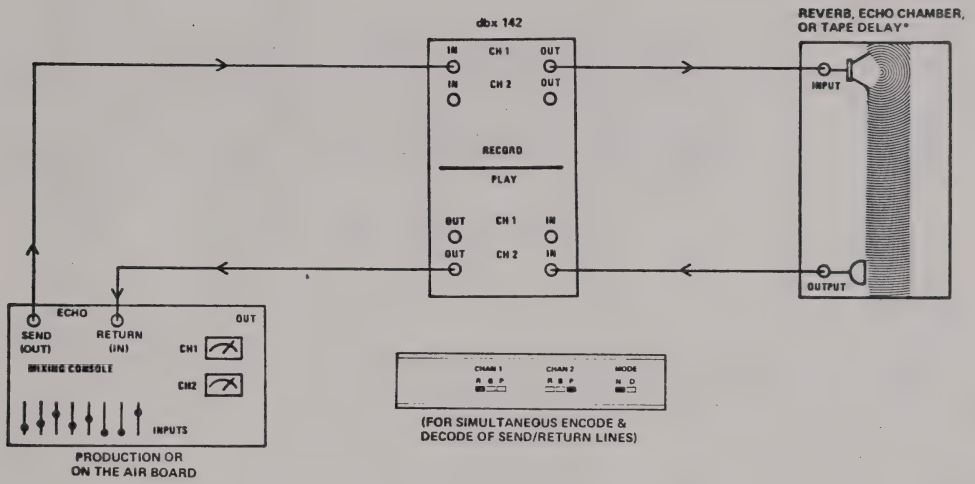
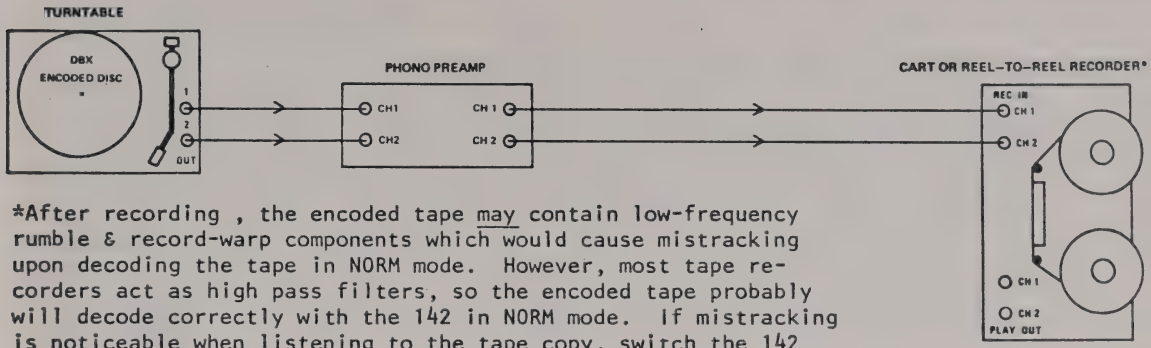


FIGURE 9. NOISE REDUCTION OF TIME DELAY DEVICES.

*Due to the expansion process that is an integral part of dbx processing, decreases in level in the storage or transmission medium are further decreased in level by decoding. This means that the decay of a live chamber, spring-type reverb, or regenerative echo device will be effectively shortened by dbx processing. Tape delay (tape record input to play or monitor output, for example) would not undergo any shortening of delay time because the output from the tape is at the same level as the input to the tape.

This effective shortening of some types of delay can be compensated for by increasing the delay time of the device, where possible. There will be a great improvement in signal quality, regardless of the delay medium.

HOW TO MAKE A TRANSFER (TAPE COPY) OF ENCODED TAPES OR DISCS



*After recording , the encoded tape may contain low-frequency rumble & record-warp components which would cause mistracking upon decoding the tape in NORM mode. However, most tape recorders act as high pass filters, so the encoded tape probably will decode correctly with the 142 in NORM mode. If mistracking is noticeable when listening to the tape copy, switch the 142 to DISC mode.

FIGURE 10A. ENCODED DISCS.

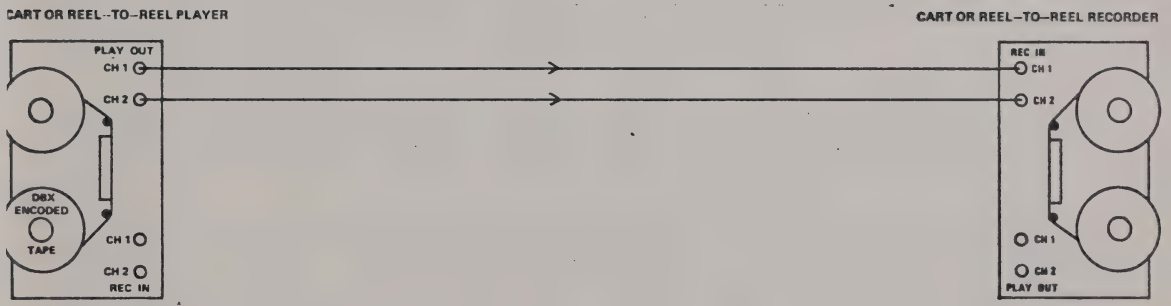


FIGURE 10B. ENCODED TAPES.

NOTE: No dbx noise reduction unit is required, thereby freeing the 142 for production. Playback levels of the tape machine or turntable with the original (master) program would be adjusted for the installation's nominal (zero VU) level, and record levels of the machines making the copy should be set for the same nominal level. (Slight differences in level will not affect the process.

SECTION IV

ADJUSTMENTS

LEVEL MATCH CALIBRATION

The dbx 142 is factory calibrated for nominal input and output levels of +4dBm. Level match trimmers are provided to maintain similar input and output levels in record and play modes, or between bypass and encode/decode processing, primarily for convenience in monitoring. Level match calibration is not necessary to maintain accurate encoding and decoding, and once set, the 142 is sufficiently stable that recalibration is seldom required. The record and play level match controls may be adjusted to accommodate nominal levels of between -10dBm and +18dBm. Bypass level match, which is only effective when the 142 is in remote bypass mode, has a smaller range of adjustment; this is because bypass is meant to have unity gain, and the only reason any adjustment may be needed is that variations in source and load impedances may cause slight loading and level differences.

Aside from the obvious monitoring convenience of having unity gain through the encode/record/play/decode process, level matching does serve a secondary purpose. It helps keep maximum levels below the clipping point of the 142 and of the circuitry which follows, just as it keeps the minimum levels above the noise floor of the 142 and of the circuitry which follows.

Due to pre-emphasis and de-emphasis, the level must be calibrated at a frequency of between 700Hz and 1kHz (no higher or lower); whatever frequency is used, it should be the same for record and play calibration. The calibration level should be equal to the nominal level (zero VU) used in your studio.

The front-panel bypass switches create a hard-wire shunt around the 142's active circuitry, so the bypass level match trimmers have no effect when these switches are engaged. However, as explained above, the trimmers do affect the level when the unit is remotely switched to bypass mode. The play and record level match trimmers affect the channel gain only when the channel is switched to the respective play or record mode.

LEVEL MATCH CALIBRATION PROCEDURE (Repeat for each channel)

1. Apply a test signal to the 142 play input terminals. (Test tones are to be from 700-1kHz, and of nominal zero VU reference level, as explained above.)

2. Place the channel in BYPASS mode, using the front panel switch. Connect a VU meter, voltmeter, or dB meter to the channel's play output terminals, and check the output level. It should be the same as the input signal level; any other reading indicates miscalibration of the meter or the tone source.

3. Switch the channel to PLAY (decode) and adjust the PLAY LEVEL MATCH trimmer, if necessary, to again obtain the same play output level as in bypass: in other words, unity decoder gain.

4. Reconnect the test signal to the record input terminals...do not change the frequency or level. Connect the meter to the corresponding record output terminals.

5. Switch the channel to RECORD (encode) mode, and adjust the RECORD LEVEL

LEVEL MATCH CALIBRATION PROCEDURE (Continued)

MATCH trimmer, if necessary, to obtain the same nominal output level: in other words, unity encoder gain.

6. With the same test signal connected to the record input, and the same meter connections on the record output, place the 142 in remote bypass mode. This may be done by grounding pins 13 and 15 of the input/output connector to any audio shield or chassis ground. The record output should simultaneously be connected to the circuit which it normally drives to obtain normal loading, if any. The front panel should still be in RECORD mode.

7. Adjust the BYPASS LEVEL MATCH trimmer to obtain the same nominal output level as in front-panel bypass mode.

This completes the level match calibration procedure. Hereafter, unless the 142 is subjected to severe conditions or to service procedures, level match calibration may be done once a year.

TAPE MACHINE ALIGNMENT

Tape machine alignment is done according to normal alignment procedure, and using any standard alignment tape. There are, however, a few things to keep in mind. Of primary importance, play and record equalization must be "just right", since frequency response errors may be increased by the dbx decoding. Record and play calibration levels should be at the established studio standard (although level match errors will not create any audible differences in the encode/decode process). In fact, the only area of alignment that may differ at all is the setting of the record bias level; typical bias settings do not correlate with minimum tape modulation noise. Therefore it is suggested that the bias settings shown in figure 11 (next page) be substituted for your usual bias. Settings are listed for most popular tapes. (Refer to the "Theory of Operation" section of this manual for additional details on tape modulation noise.)

TAPE MACHINE ALIGNMENT PROCEDURE (Repeat for each channel)

1. Place a standard alignment tape on the machine and perform routine playback alignment, including equalization and azimuth. Use the tape machine VU meters, or place the 142 in BYPASS mode and use the console VU meters. Set play level for the nominal studio reference level (0 VU). Remove the alignment tape.

2. With the 142 in BYPASS mode, apply a test tone to the 142 record input (700Hz to 1kHz, at studio reference level). Set the record bias in accordance with the values suggested in figure 11 (next page). Then set the record calibrate level to obtain a zero VU indication on the tape machine's record input meter. Record about 30 seconds at the calibrated level on a length of blank tape.

TAPE MACHINE ALIGNMENT PROCEDURE (Continued)

3. Apply a 7kHz to 10kHz tone to the record input. On 3-head machines, switch the VU meters to monitor (off tape) mode and adjust the azimuth for maximum (peak) output. With 2-head machines, the setting of the play azimuth (with an alignment tape) automatically sets the record azimuth, so this step is not required.

4. Adjust the record frequency response trimmers, according to the manufacturer's instructions, and again check the record calibrate level at 700Hz to 1kHz.

   NOTE: Steps 1-4 are representative of a "typical" play/record alignment, but you are free to substitute whatever alignment procedure you are currently using.

5. Place the 142 in RECORD mode, apply a 700Hz to 1kHz, 0 VU tone to its record input, and record a 30 second test. Rewind the tape, switch the 142 to PLAY mode, and play the test recording. This test tone should be at the same level as the original record input tone, measured at the 142 play output terminals, play input terminals, or anywhere in the record/encode/play/decode process.

This completes the tape machine alignment procedure. It has been assumed that the 142 level match controls are properly adjusted, as outlined in the preceding portion of this section. If level match is off, the check performed in step #5 (above) may show a discrepancy in level. If step #5 shows a discrepancy, and the level match is properly set, then the record or play calibrate levels of the tape machine are improperly set. Remember, the actual levels are not critical, but are matched basically for monitoring convenience.

FIGURE 11. RECORD BIAS SETTINGS FOR MINIMUM TAPE MODULATION NOISE.

SECTION V
OPERATION

HINTS FOR OPTIMUM UTILIZATION OF THE 142

1. Except for record/playback, time delays, or gross (linear) gain changes, DO NO SIGNAL PROCESSING OF DBX-ENCODED AUDIO. Encoded signals must be decoded before being mixed, limited, equalized, or otherwise processed. Naturally, signals which have been mixed or otherwise processed after decoding may again be dbx encoded.

2. When recording, USE LIMITING AND COMPRESSION FOR EFFECTS ONLY. The dbx encoding process makes routine limiting and compression unnecessary in the studio.* (As stated above, any such signal processing would be done before encoding.)

3. A PROGRAM THAT HAS BEEN ENCODED ON ONE DBX CHANNEL MAY BE DECODED ON THAT OR ANY OTHER CHANNEL. Tight manufacturing tolerances assure proper tracking without need for special alignment.

4. FOR TRANSFERS (DUBS), ENCODED TAPES & DISCS NEED NOT BE DECODED AND RE-ENCODED. The encoded program may be played directly into the record input of the machine making the copy. Of course a tape may be decoded, if desired, then re-encoded . . . many times, and without serious side effects. Either way, numerous generations of copies can be made with no increase of tape noise. (These procedures are based on real-time copying at the original tape speed; for high speed duplication, consult the dbx factory.)

5. Remember that DBX PROCESSING CUTS THE DELAY TIME OF A LIVE CHAMBER OR REVERB to approximately half the unprocessed time, but the quality is excellent.

6. ADJUST THE TAPE RECORDER BIAS FOR MINIMUM MODULATION NOISE. As explained in the "Theory of Operation" section of this manual, the bias required for minimum modulation noise may differ slightly from standard record bias. Suggested bias settings are shown in figure 11, in the "Tape Machine Alignment" section of this manual.

7. IT IS NOT NECESSARY TO INCREASE RECORD LEVELS TO "TAKE ADVANTAGE" OF ELEVATED LEVEL TAPES. Instead, it is usually desirable to retain standard record and play levels. This provides extra headroom for truly accurate transient response and less chance of tape saturation. The extra few dB of recorded level are not really necessary because dbx processing avoids residual tape noise completely, even with standard tapes.

8. Do not be alarmed when the tape machine VU meters do not track the console VU meters. 2:1 compression and pre-emphasis in the 142 level-sense circuitry will cause this effect. Therefore, USE THE CONSOLE VU METERS, AND DISREGARD THE TAPE MACHINE METERS.

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*Limiting at the transmitter, after decoding, may still be necessary in order to satisfy FCC regulations.

NOTE: A variety of other dbx professional noise reduction equipment is available, including 4-track and larger units intended for making noise-reduced multi-track master tapes. In additon, dbx manufactures a full line of consumer noise reduction equipment. For information concerning any of this equipment, as well as information on the manufacturing and availability of dbx-encoded discs, contact the factory or your local dbx dealer.

THEORY OF OPERATION

Two basic approaches to noise reduction are in current use. One removes noise from previously recorded material in which no form of noise reduction was used. The other applies signal processing at the time of original recording, and again at playback. As explained in the beginning of this manual, dbx noise reduction involves the latter approach.

An idealized tape noise reduction system can be achieved using compression before recording and expansion upon playback. This is an old idea, but it has been difficult to apply to professional quality audio systems. The system's compressor and expander must track each other accurately; that is, their action must be precisely complementary, especially on transient signals. The problem with achieving truly complementary action is that of accurately detecting the signal level.

Which method of signal level detection is used?

There have been many attempts to produce a commercially feasible compression/expansion noise reduction system. Methods used in these systems include (1) peak and average detection for level sensing to control the compressor and expander, (2) dividing the frequency spectrum into bands to achieve reasonable accuracy in encode/decode tracking, (3) elaborate setup and alignment procedures requiring pilot tones, and (4) limitation of the amount of noise reduction used so that the remaining hiss is sufficient to mask imperfections in the system.

RMS detection is considered the ideal means of signal detection because it permits accurate compressor/expander tracking, regardless of phase differences contributed by a tape recorder (or by transmission lines). RMS represents the sum of the individual frequency energies present in the signal, without regard to phase relationships. Until now, RMS detection has been very complex and very expensive.

Evolution of the dbx system design

A linear decibel compressor/expander may be created in the manner illustrated by figure 12 (next page). Voltage Controlled Amplifiers (VCA's) control the input and output gain. Input and output level sense detectors control the VCA gain with an appropriate polarity to produce either compression (Fig. 12-A) or expansion (Fig. 12-B). Such a system might seem practical as a noise reduction measure; using a 2:1 compression/1:2 expansion factor, for example, the dynamic range could be doubled. Thus, a 60dB S/N recorder could capture a 120dB S/N program. Unfortunately, *the instantaneous background noise level varies with the signal envelope.*

The human ear can "tune out" a steady hiss to the point where the listener is only subliminally conscious of it, but a varying noise envelope has a much higher auditory annoyance factor. Consequently, the system in figure 12 is quite unsatisfactory for critical applications, despite its apparent outstanding performance on paper.

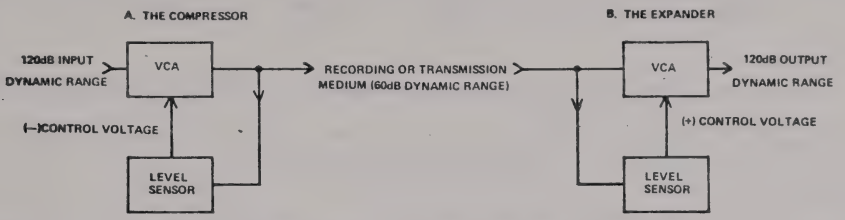


FIGURE 12. A LINEAR COMPRESSOR/EXPANDER SYSTEM USING VCA's. It looks effective on paper, but the failure to reduce tape modulation or asperity noise makes it unnacceptable for critical applications.

Tape noise...where it comes from

It is important to understand the behavior of the noise sources in tape recording. We are all familiar with the sound of the background noise in the absence of any signal. The background noise spectrum typically increases in level as the recorded signal level increases. In the case of a single-frequency tone, as the signal level is raised, the broadband noise increases due to the increased number of magnetized domains passing by the playback head gap. At the same time, the spectral distribution is such that an even greater noise build-up occurs near the frequency of the tone; this is due to inhomogeneity in magnetic particle distribution in the tape coating, and to surface asperities which lift the tape away from the gap as they pass by. Both these mechanisms (more domains passing the gap, and particle/surface deviations) cause a random-function amplitude modulation of the recorded signal. This, in turn, results in noise sidebands which create a fairly constant, instantaneous signal-to-noise ratio for recorded levels above -10dB. Below -10dB, the spectral distribution decreases in such a way that the noise sidebands are "buried" in the minimum residual broadband noise; the effect is a decrease in the instantaneous signal-to-noise ratio at lower recorded levels. Thus, the noise sidebands, which drop in level as the frequency departs from the recorded tone, quickly become masked by the residual noise.

With most audio signals, masking effects created by the recorded signal cover up noise effects throughout several octaves on either side of the frequency band containing the dominant energy. However, background hiss which is about 50dB under a strong low-frequency signal (typical) can be plainly heard at higher frequencies, where masking effects cease to function. The result is an apparent increase in the high frequency content of tape recorded signals. (This is why, at first, dbx noise reduction may seem to reduce high frequencies. An actual A-B comparison of live program with program that is dbx encoded and decoded will show that there is actually no difference in frequency response.)

NOTE: For further information on the effects of noise in recording, refer to these articles: "The Measurement and Subjective Assessment of Modulation Noise in Magnetic Recording," E.G. Trendell, Journal of the Audio Engineering Society, Vol. 17, #6, Dec. 1969; "Masking of White Noise by Pure Tone, Frequency-Modulated Tone and Narrow-Band Noise," I.M. Young and C.H. Wenner, Journal of the Acoustical Society of America, Vol. 41, #3, 1967.

Using signal pre-emphasis and de-emphasis to overcome noise

To minimize the modulation or asperity noise which can be heard with strong, low-frequency signals, a high frequency pre-emphasis is applied prior to compression (in the encoding process). Reciprocal de-emphasis is then applied after expansion (in the decoding process). This maintains a constant frequency response for higher frequency signals that were in the original program, but high frequency noise that develops on the tape, being fairly low in level, is expanded to inaudibility by the de-emphasis (upon decoding).

FIGURE 13. A BASIC APPROACH TO PRE-EMPHASIS AND DE-EMPHASIS.
For reduction of tape modulation or asperity noise.

Using pre-emphasis of the level sensor (detector) to avoid tape saturation

If a gradual signal weighting curve is used, as shown in figure 14, then the high frequency hiss that accompanies a strong low-frequency signal is reduced by the maximum amount of weighting (16dB in this case). This helps to eliminate what is commonly known as "pumping" or "breathing." However, there will be a sharp increase in susceptibility to high-frequency self erasure...already a problem without pre-emphasis with many types of signals found in contemporary music. In addition, this large amount of pre-emphasis can lead to high-frequency tape saturation.

To prevent tape saturation and self-erasure, pre-emphasis is also added to the RMS level detector. Since any desired frequency-shaping function may be used on the detector (so long as it is the same for record and play pre-emphasis), the detector pre-emphasis which has been chosen is not identical to the signal pre-emphasis. Essentially, pre-emphasis of the RMS detector in the encoding process prevents any signal pre-emphasis from raising the high-frequency component fed to the recorder beyond "safe" limits. Identical pre-emphasis of the RMS detector upon playback has a reciprocal effect, due to the reversed polarity of the detector in this mode, thereby maintaining the correct mirror-image decoding (refer to figures 15 and 16).

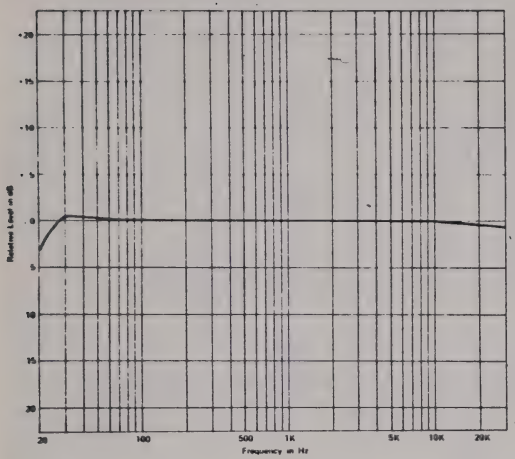


FIGURE 14-A. SIGNAL BANDPASS FILTER AVOIDS MISTRACKING DUE TO VERY HIGH AND VERY LOW FREQUENCY RECORD-INPUTS WHICH CANNOT BE ACCURATELY RECORDED, BUT WHICH COULD OTHERWISE ACTIVATE THE LEVEL-SENSING CIRCUITRY.

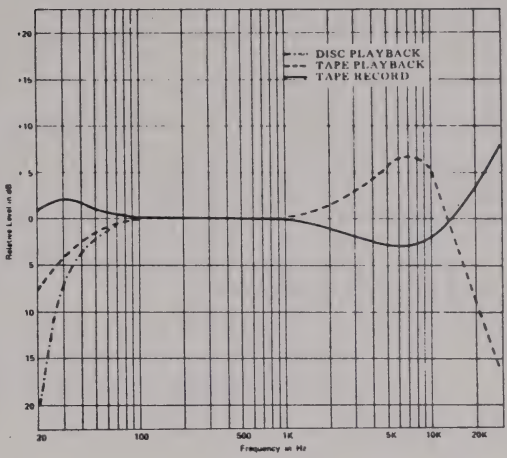


FIGURE 14-B. FREQUENCY RESPONSE CURVES FOR RECORD (ENCODE) AND PLAY (DECODE) PROCESSING. THESE SINGLE-FREQUENCY RESPONSE CURVES ILLUSTRATE HOW THE DISC PLAYBACK IS SHARPLY ATTENUATED BELOW 30Hz, WHEREAS THE TAPE (NORM) PLAYBACK ROLLS OFF SLIGHTLY. (LARGE APPARENT LOSS AT 30Hz IS OFFSET BY PEAK IN RECORD RESPONSE AT 30Hz.)

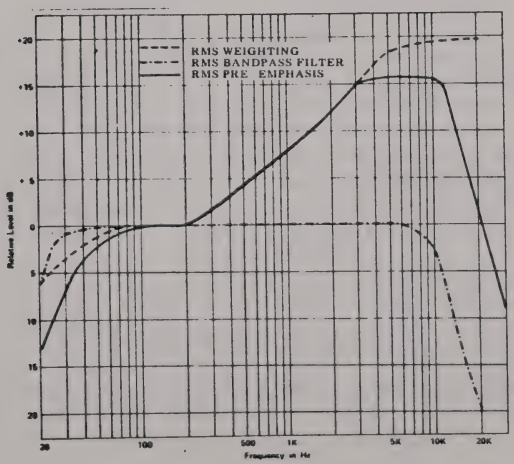


FIGURE 15. RMS LEVEL-SENSE DETECTOR PRE-EMPHASIS. RMS WEIGHTING PLUS DETECTOR BANDPASS FILTER YIELD RMS PRE-EMPHASIS (SOLID LINE). NOTE THE DIFFERENCE BETWEEN THIS AND THE SIGNAL PRE-EMPHASIS (FIG.14B).

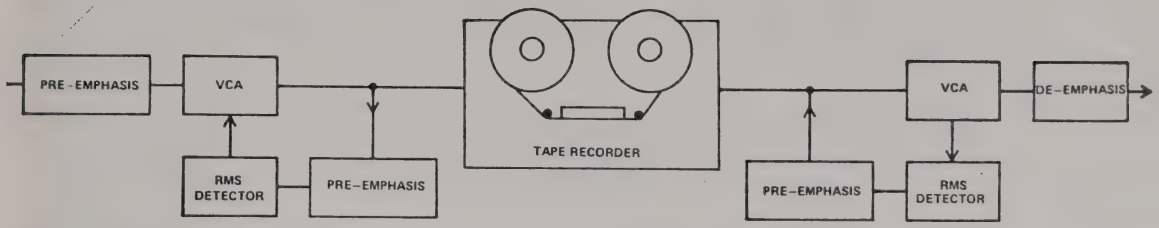
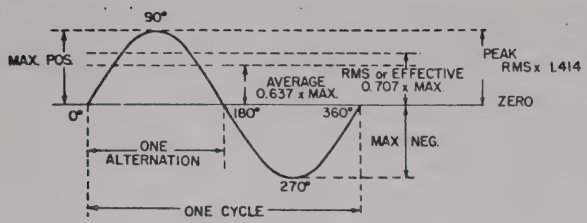


FIGURE 16. A DETAILED APPROACH TO PRE-EMPHASIS AND DE-EMPHASIS, WITH ADDITIONAL DETECTOR PRE-EMPHASIS TO AVOID EXCESSIVE RECORD LEVELS AT HIGHER FREQUENCIES.

Using RMS detection to avoid phase sensitivity and to maintain excellent transient response.

A system using pre-emphasis and de-emphasis, along with compansion, is useful only if it has adequate transient response tracking properties—especially when there is a tape recorder between the compressor and expander. Peak responding level sensors seem, at first glance, ideal. Unfortunately, peaks often occur which are as high as 24dB above VU meter readings (VU meters are averaging devices). Thus, tape recording levels are commonly run in a region where such peaks are "flattened" by tape saturation. This peak truncation is gentle, but would nevertheless create tracking errors between record and play level sensors which function according to peak detection. For that matter, the peak truncation would cause problems with any level sensor that is phase sensitive. Moreover, averaging level sensors would be too slow, allowing brief transients to create mistracking. Most tape recorders and remote lines have considerable time-delay dispersion (frequency-dependent phase shifts that do not have constant time delay characteristics). Therefore, neither the peak nor the average value of a signal is easily recovered from the storage or transmission medium, but one function does remain unchanged and can be recovered—the RMS value of the signal.

The RMS value is the sum of the energy of all frequency components present, without regard to their phase relationships. (Actually, RMS—Root Mean Square—is the square root of the sum of the squares of the individual energies.) Recently developed analog techniques make it possible to sum the squares of the instantaneous signal levels, so RMS determination is feasible; in fact, it is the basis of all dbx noise reduction systems. The relationship between peak, average, and RMS values is illustrated in figure 17, on the next page.



rms = 0.707 × peak voltage
rms = 1.11 × average voltage
peak = 1.414 × rms voltage

FIGURE 17. PEAK, AVERAGE, AND RMS VALUES OF A SIMPLE SINE-WAVE SIGNAL. WHILE ALL THESE MEASUREMENTS ARE RELATIVELY SIMPLE FOR PURE SINE WAVES, RMS IS THE ONLY VALUE WHICH CAN BE RELIABLY RECOVERED AFTER RECORDING OR TRANSMISSION OF A COMPLEX SIGNAL.

Using bandpass filters to avoid mistracking

It is possible for an expander to mistrack due to subsonic signals (such as room vibration from a passing train) or ultra-sonic signals (such as record bias). To prevent such mistracking, bandpass filters are placed in the signal path, and also in the level-sense detection path of the 142. The bandpass filter used in the dbx II signal path is 20Hz-20kHz, with a slight roll-off below 30Hz. Thus, the units pass the full useable audio spectrum without difficulty.

A signal detector, however, needs to operate only in a frequency range where the dominant signal energy exists. Therefore, the bandpass filter for the level-sense detector passes frequencies between 30Hz and 10kHz. This allows live recording without errors due to passing trucks, subways, ventilating fans, etc. The bandpass also helps eliminate mistracking when recording tapes from warped phonograph records.(For playing dbx-encoded discs, DISC mode further modifies the low frequency signal cutoff to prevent record warp or rumble from creating mistracking of the expander.) The sometimes unavoidable roll-off in the high-frequency response of cart recorders, or of land lines and full-frequency phone lines, is similarly prevented from causing tracking errors...the detector sensitivity is cut off at 10kHz. At the same time, the presence of too much high-frequency information, such as spurious oscillations or leakage of record bias sub-harmonics, is blocked from the level sense detector. (Remember, the overall encode/decode frequency response is still 30Hz-20kHz, ±1dB.)

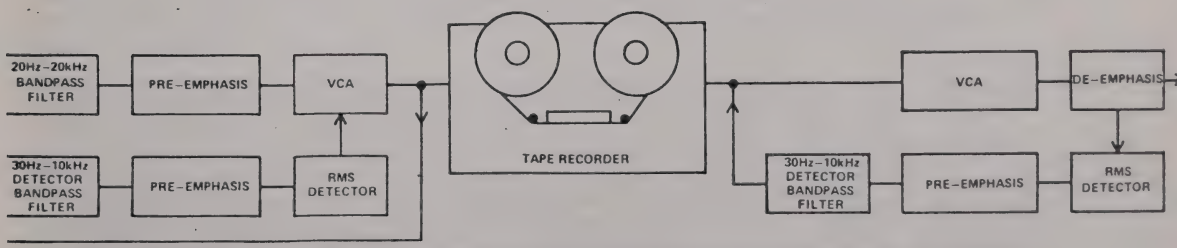


FIGURE 18. BANDPASS FILTERS PREVENT MISTRACKING OF GAIN CONTROL CIRCUITRY.

SYSTEM PERFORMANCE

The original dbx noise reduction system has become the newly accepted technical standard in the recording industry, and is used for noise reduction in many recording studios throughout the world. dbx optimized this original system for broadcast applications, and the result is known as dbx II. dbx II is nearly identical to the original system, offering the same amount of noise reduction and the same headroom improvement. However, differences in the pre-emphasis and de-emphasis curves, as well as in the level-sense bandpass, make it inadvisable to interchange recording studio dbx-encoded tapes with dbx II decoding, or vice-versa.

The dbx II system, as packaged in the Model 142, has an equivalent "A" weighted background noise of under -90dBm...without audible imperfections (when used with a studio tape recorder). A difference in the apparent high-frequency response may be heard as the 142 is alternately switched in or bypassed while recording and simultaneously monitoring the playback. However, careful comparison of the original signal with both the encoded/decoded and with unprocessed recorded/played audio will show that the noise-reduced signal is virtually identical to the original signal. The difference in apparent high frequency response is due to increased tape hiss that is present in the unprocessed recordings, hiss which has been interpreted by the ear as a "hotter" top end.*

There are some barely audible imperfections, even after dbx processing, which are due to asperity noise with some types of very clear signals. Unfortunately, these noise components lie in the same frequency region as the dominant signal energy band, hence we must look to improvements in tape surfaces for their reduction...regardless of the noise reduction system in use. Incidentally, these asperity noise components can be masked, almost completely, by a steady hiss. The required level of this "noise perfume" is about -65dBm. It is no coincidence that noise reduction systems which claim "absolutely no audible effect" have this residual noise present in the output (dbx noise is below -90dBm).

By taking advantage of dbx noise reduction wherever possible, from the original recording right up to the transmitter input, the quality of your audio signal will be greatly improved. Not only will there be less hiss, giving a better signal-to-noise ratio (and improved coverage), there also will be a much more "live" feel to the programs, whether live or taped. Moreover, the distortion that would otherwise have been introduced by tape saturation, excessive limiting and/or transient overloads will not act to degrade the signal quality. Once you've tried dbx noise reduction, you'll never settle for less.

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*People who are not accustomed to hearing live performances on studio monitor speakers are used to hearing hiss as part of the program. When this hiss is removed by dbx processing, the program may appear to have lost some top end. However, an "A-B" comparison of the program, (A) before recording and (B) after dbx-encoding/recording/playback/decoding, will immediately convince the listener that there is actually no loss of high frequency response. Furthermore, an "A-B" of unprocessed and dbx-processed recordings will show that hiss and distortion are far greater in the unprocessed recording.

SECTION VI
APPENDIX

The following pages are schematic and assembly diagrams of the 142. They have been provided for the purpose of enabling a better understanding of the unit's function. Service should be referred to qualified personnel.

Connectors of the type used on the 142 rear panel are available from Cinch Jones and Vernitron.* The part numbers are as follows:

	Cinch Jones P/N	Vernitron P/N
Male Connector	P-327-CCT	P-3327-04-CCT
Female Connector	S-327-CCT	S-3327-04-CCT

The Vernitron connectors may be ordered with gold-plated contacts; Cinch Jones connectors are available only with cadmium-plated contacts.

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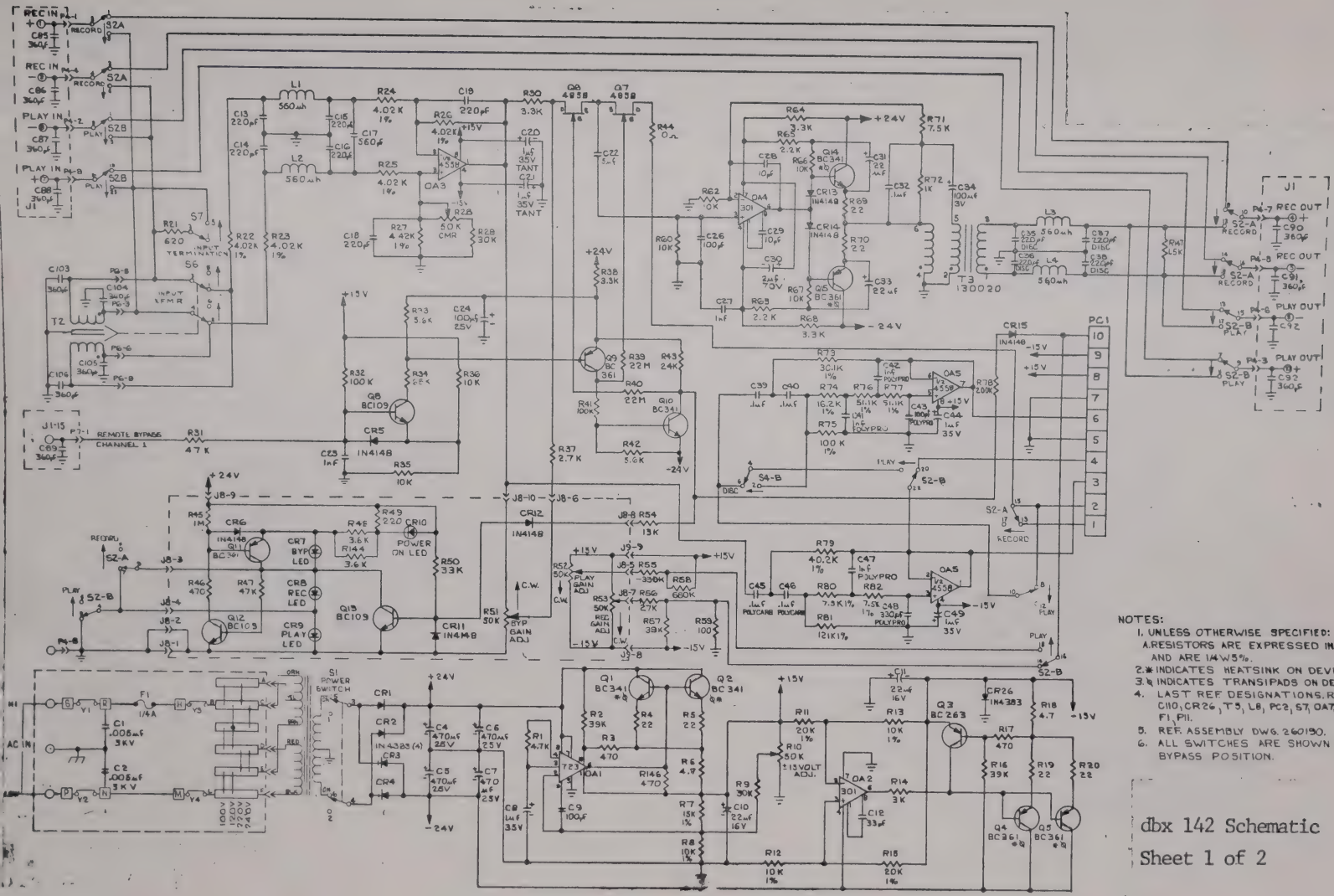
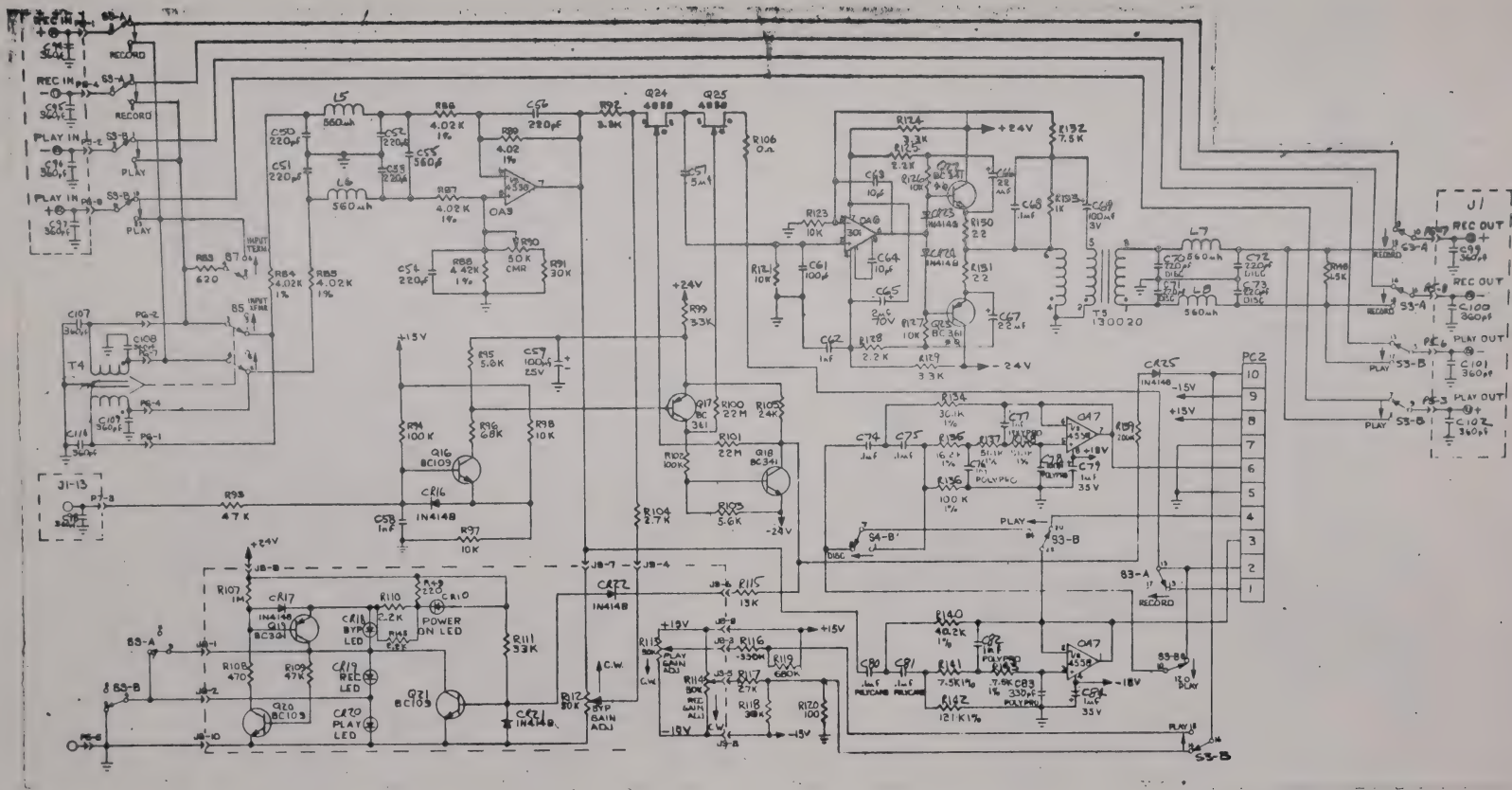
*Vernitron Electrical Components, 117 Union Avenue, Laconia, NH 03246. Phone (603) 524-5101, TWX 710-364-1843.



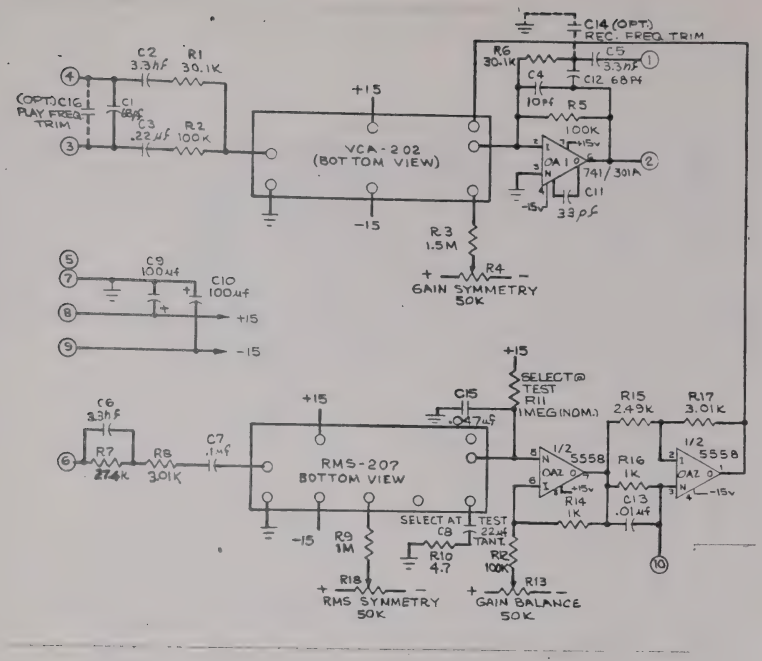
dbx 142 Schematic

Sheet 1 of 2

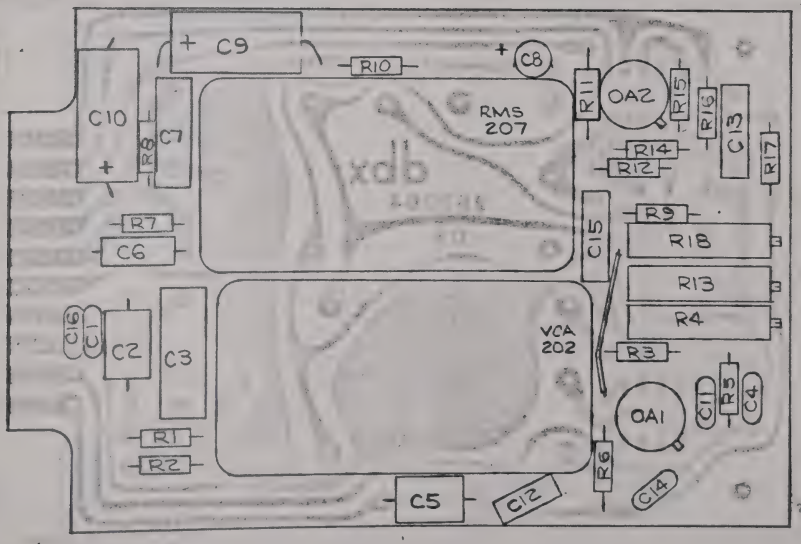


dbx 142 Schematic

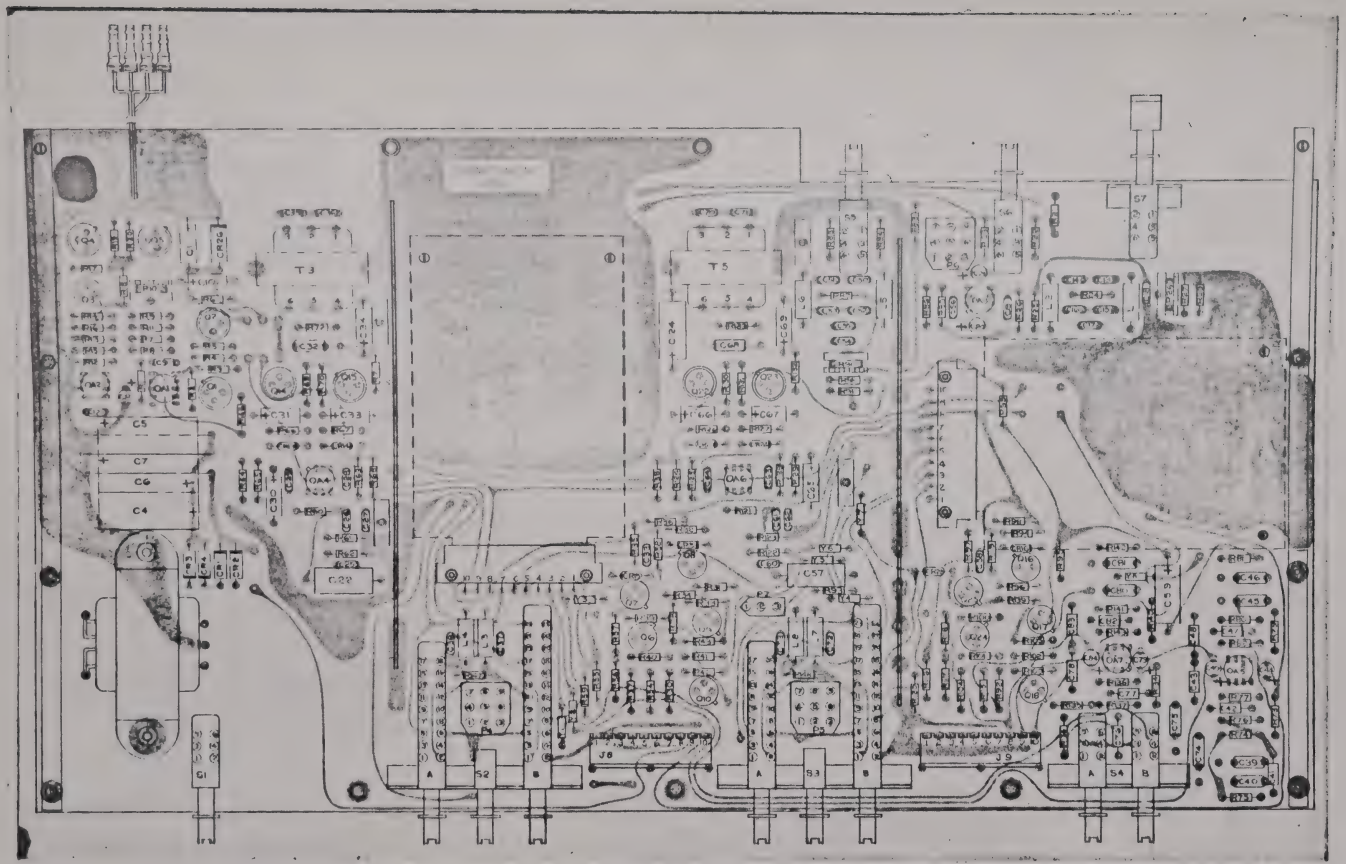
Sheet 2 of 2



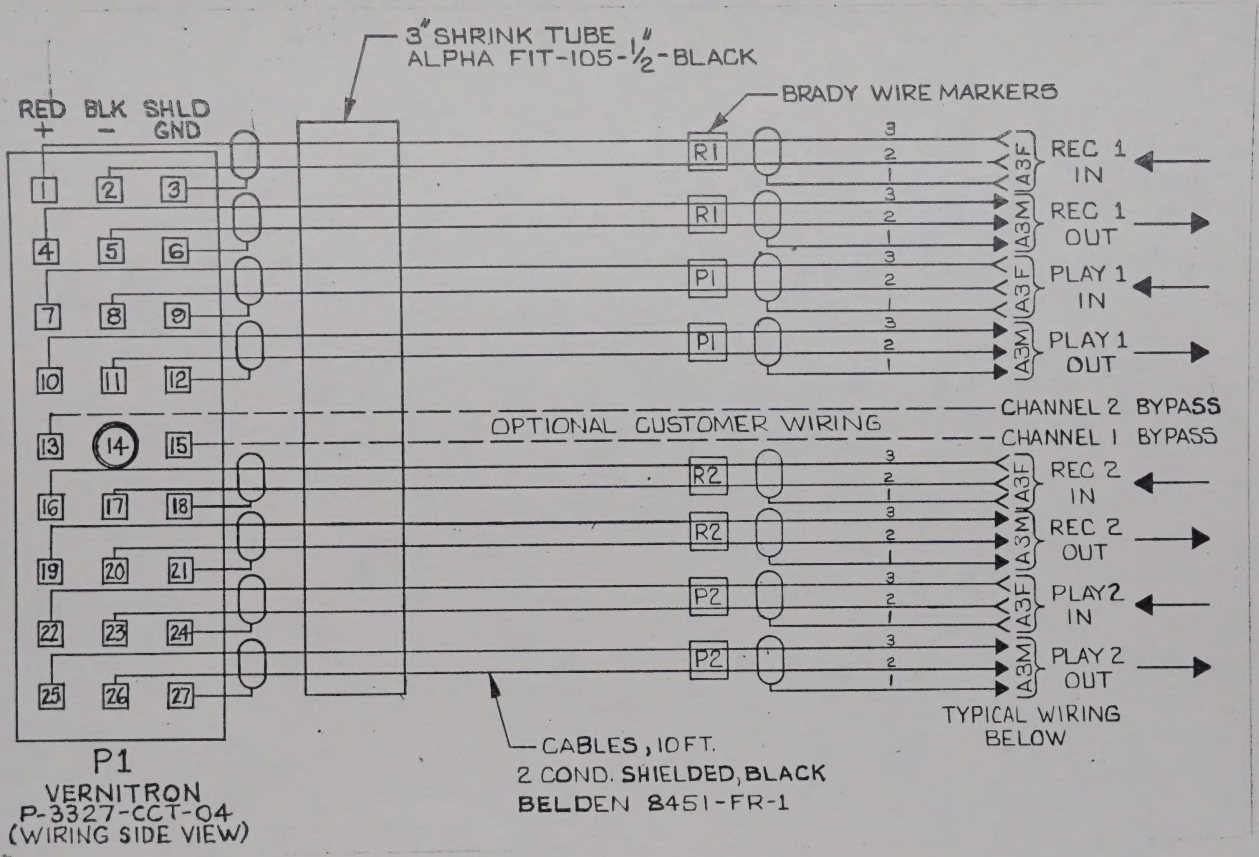
dbx 304 Schematic



dbx 304 Parts Layout



dbx 142 Parts Layout



dbx 142 Cable Wiring Diagram

ACCEPTED